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EXAMINER	
GODBOLD, DOUGLAS	

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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No. 10/695,125	Applicant(s) SINGHAL, MANOJ	
	Examiner DOUGLAS C. GODBOLD	Art Unit 2626	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 12 August 2008.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-6,8-24 and 27* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-6,8-24 and 27* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

1. This Office Action is in response to correspondence filed August 12, 2008 in reference to application 10/695,125. Claims 1-6, 8-24 and 27 are pending and have been examined.

Response to Amendment

2. The amendments filed August 12, 2008 have been accepted and considered in this office action. Claims 1, 3-6, 8, 14, 16, 18, and 19 have been amended, claims 7, 25, and 26 cancelled, and claim 27 has been added.

Response to Arguments

3. Applicant's arguments with respect to claims 1-6,, 8-24 and 27 have been considered but are moot in view of the new ground(s) of rejection. The previous rejections under 35 U.S.C. 112 have been withdrawn.

Claim Rejections - 35 USC § 112

4. The following is a quotation of the first paragraph of 35 U.S.C. 112:

> The specification shall contain a written description of the invention, and of the manner and process of making and using it, in such full, clear, concise, and exact terms as to enable any person skilled in the art to which it pertains, or with which it is most nearly connected, to make and use the same and shall set forth the best mode contemplated by the inventor of carrying out his invention.

5. Claims 6, 19, and 27 are rejected under 35 U.S.C. 112, first paragraph, as failing to comply with the written description requirement. The claim(s) contains subject matter which was not described in the specification in such a way as to reasonably convey to

one skilled in the relevant art that the inventor(s), at the time the application was filed, had possession of the claimed invention. Claim 6 and 19 are directed towards decimation of frequency components of audio signals. However, the previous versions of the claims and the specification only support decimation of time domain signals. Because of this, for purposes of examination, it will be interpreted that frequency components are converted to time components and then decimated. Claim 27 is directed to selected audio components above a certain frequency. However, this is not described in the original specification or claims.

Claim Rejections - 35 USC § 103

6. The text of those sections of Title 35, U.S. Code not included in this action can be found in a prior Office action.

7. Claims 1, 3, 4, 10, 11, 13, and 14 are rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders (Real-Time Discrimination of Broadcast Speech/Music) in view of Tzanetakis et al (Sound analysis Using MPEG Compressed Audio).

8. Consider claim 1, Saunders teaches a method for classifying an audio signal (we describe a technique which is successful at discriminating speech from music; page 993, column 1, line 1), the method comprising:

receiving an audio signal to be classified (this is a technique for discriminating speech from music from an FM broadcast; page 993, column 1, line 2);

analyzing selected audio signal components (The first step is to measure the ZCR of the signal over a 2.4 second segment of the data; page 994, column 2, line 43);

recording a result of analysis of the selected audio signal components (would be inherent in order to compare it);

comparing the recorded result of analysis to a threshold value (If this statistic exceeds a specific threshold, the distribution outside these bounds is significantly skewed and the waveform is likely speech; page 994, column 2, line 43); and

classifying the audio signal based upon comparison of the recorded result of analysis and the threshold value (If this statistic exceeds a specific threshold, the distribution outside these bounds is significantly skewed and the waveform is likely speech; page 994, column 2, line 43).

Saunders does not specifically teach that the audio signal components are audio frequency components.

In the same field of audio analysis, Tzanetakis teaches the audio signal components are audio frequency components. (Tzanetakis analyzes MPEG audio files, which stores subband information that were converted via a filter bank; overview of MPEG 2. IN order to use traditional analysis such as zero-crossing, MPEG data must be decoded; introduction 1, paragraph 2).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to combine the decompression of Tzanetakis with system of Saunders in order to be able to apply the traditional methods of Saunders to MPEG audio files, Cook introduction 1, paragraph 2.

9. Consider claim 3, Saunders and Tzanetakis teach the method according to claim 1, wherein analyzing the selected audio frequency components comprises transforming the selected audio frequency components to time domain components (Tzanetakis analyzes MPEG audio files, which stores subband information that were converted via a filter bank; overview of MPEG 2. decoding MPEG inherently comprises transforming subband information back to time domain signals) and counting zero point transitions of the selected audio signal components (The first step is to measure the ZCR of the signal over a 2.4 second segment of the data; page 994, column 2, line 43. Measuring the Zero Crossing Rate would entail counting the number of zero crossings).

10. Consider claim 4, Saunders and Tzanetakis teach the method according to claim 1, wherein recording a result of analysis of the selected audio frequency components comprises transforming the selected audio frequency components to time domain components (Tzanetakis analyzes MPEG audio files, which stores subband information that were converted via a filter bank; overview of MPEG 2. decoding MPEG inherently comprises transforming subband information back to time domain signals) and recording a count value of a number of zero point transitions of the selected audio signal components (The first step is to measure the ZCR of the signal over a 2.4 second segment of the data; page 994, column 2, line 43. Measuring the Zero Crossing Rate would entail counting the number of zero crossings. This number would inherently have to be stored somewhere in order to process it or manipulate it).

11. Consider claim 10, Saunders teaches the method according to claim 1, wherein classifying the audio signal occurs at a receiving end of an audio transmission system (this is a technique for discriminating speech from music from an FM broadcast; page 993, column 1, line 2).

12. Consider claim 11, Saunders teaches the method according to claim 1, wherein the audio signal is one of an analog signal and a digital signal (A sample rate of 16Khz was chosen for this discrimination technique; page 995, column 1 line 1. If something is sampled it is well understood that it is being converted to a digital signal. this is a technique for discriminating speech from music from an FM broadcast; page 993, column 1, line 2. This further tells us that the signal started out as an analog signal as at the time of the publication of Saunders all FM broadcasts were analog.).

13. Consider claim 13, Saunders teaches the method according to claim 1, wherein the threshold value used in the comparison determined through trial and error of a plurality of iterations in a comparing device (Data was collected manually by listening, collecting and storing features, and labeling the segment. A variety of content was processed, including talk, commercials, and many types of music. Once the classifier was trained, the parameters were stored and fed into the real-time feature extraction/classifier routine; page 995, column 1, line 33).

14. Consider claim 14, Saunders teaches the method according to claim 1, wherein analyzing selected audio frequency components comprises counting zero point transitions of the audio signal for a predetermined period of time (The first step is to measure the ZCR of the signal over a 2.4 second segment of the data; page 994, column 2, line 43. Measuring the Zero Crossing Rate would entail counting the number of zero crossings).

15. Claim 2 is rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders in view of Tzanetakis as applied to claim 1 above, and further in view of Carey (A Comparison of Features for Speech, Music Discrimination).

16. Consider claim 2, Saunders in view of Tzanetakis teaches the method according to claim 1, but does not specifically teach wherein classifying the audio signal based upon comparison of the recorded result of analysis and the threshold value further comprises:

if the recorded result of analysis is greater than the threshold value, then the audio signal is determined to be music; and

if the recorded result of analysis is less than the threshold value, then the audio signal is determined to be speech.

In the same field of speech/music discrimination, Carey teaches if the recorded result of analysis is greater than the threshold value, then the audio signal is determined

to be music (table 1 shows that the mean value of number of zero crossing (u) for music 0.18 is greater than that of speech 0.17); and

if the recorded result of analysis is less than the threshold value, then the audio signal is determined to be speech (table 1 shows the mean value of zero crossing for speech 0.17 was less that music 0.18).

Although Saunders in view of Tzanetakis uses a slightly different zero crossing analysis method than does Carey, it would have been obvious to one of ordinary skill in the art at the time of the invention to use the parameters of Carey as this method would be computationally inexpensive (Carey page 151, column 2, section 4.4).

17. Claims 5, 16, and 18-24 are rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders in view of Tzanetakis as applied to claim 1 above, and further in view of Pohlmann (Principles of Digital Audio).

18. Consider claim 5, Saunders teaches the method according to claim 1, further comprising selecting audio frequency components prior to analyzing selected audio frequency components (The first step is to measure the ZCR of the signal over a 2.4 second segment of the data; page 994, column 2, line 43. Inherent the segment is selected first), but does not specifically teach

wherein said selecting audio signal components comprises passing the audio signal through a low pass filter for filtering out audio frequency components having a

frequency greater than a predetermined frequency thereby reducing an amount of audio information to be analyzed

In the same field of audio analysis, Tzanetakis teaches transmitting an audio signal using MPEG encoder that samples at a rate of 22050Hz; Section 3.

Therefore it would have been obvious to combine the sampling of audio for transmitting of Tzanetakis with the classification system of Saunders in order to be able to apply the traditional methods of Saunders to MPEG audio files, Cook introduction 1, paragraph 2.

This does not teach specifically wherein said selecting audio signal components comprises passing the audio signal through a low pass filter for filtering out audio signal components having a frequency greater than a predetermined frequency thereby reducing an amount of audio information to be analyzed.

In the same field of audio encoding, Pohlmann suggests passing the audio signal through a low pass filter for filtering out audio signal components having a frequency greater than a predetermined frequency thereby reducing an amount of audio information to be analyzed (sampled audio must be passed through a low pass filter at the Nyquist frequency in order to prevent distortion called aliasing; page 30, prevention section. By Nyquist filtering, signal components have been reduced.).

Therefore it would have been obvious to combine the sampling of Tzanetakis with the filtering of Pohlmann in order to prevent aliasing, and to provide a way to digitize the audio signal for analysis, coding and transmission.

19. Consider claim 16, Saunders teaches an apparatus for classifying an audio signal (The experimental setup used a Gradient A/D unit attached to a workstation; page 995, column 1, line 38), the apparatus comprising:

a zero point counter for counting and recording zero point transitions encountered in analysis of the selected audio signal components (The first step is to measure the ZCR of the signal over a 2.4 second segment of the data; page 994, column 2, line 43); and

a comparator for comparing a recorded result of analysis to a threshold value and classifying the audio signal based upon comparison of the recorded result of analysis and the threshold value (If this statistic exceeds a specific threshold, the distribution outside these bounds is significantly skewed and the waveform is likely speech; page 994, column 2, line 43).

Saunders does not specifically teach that the audio signal components are audio frequency components.

In the same field of audio analysis, Tzanetakis teaches the audio signal components are audio frequency components. (Tzanetakis analyzes MPEG audio files, which stores subband information that were converted via a filter bank; overview of MPEG 2. IN order to use traditional analysis such as zero-crossing, MPEG data must be decoded; introduction 1, paragraph 2).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to combine the decompression of Tzanetakis with system of Saunders

in order to be able to apply the traditional methods of Saunders to MPEG audio files, Cook introduction 1, paragraph 2.

However Saunders does not specifically teach a circuit for packetizing the audio signal into packets, said packets including a header, said header including a flag indicating classification of the audio signal.

In the same field of music and speech discrimination Benyassine teaches a circuit for packetizing the audio signal into packets (The encoder 112 segments the digitized speech signal into frames to generate a bitstream. In one embodiment, the speech coding system 100 uses frames having 160 samples and corresponding to 20 milliseconds per frame at a sampling rate of about 8000 Hz. The encoder 112 provides the frames via a bitstream to the communication medium 104; column 3, line 56. A window is the same as a packet in this use in the communication medium. communication devices 102 and 106 may be cellular telephones radios, or VoIP systems; column 3 line 6-11. Cell phones and VoIP systems both used packetized data), said packets including a header, said header including a flag indicating classification of the audio signal (all flags used to mark audio frames are shown in Table 1, column 9. The music detection flag F_M is set if either threshold for music conditions are met; column 7 line 37).

This combination of Saunders and Benyassine does not specifically teach at least one audio signal component reducer for selecting a reduced number of audio signal components for analysis.

In the same field of audio processing, Pohlmann teaches, at least one audio signal component reducer for selecting a reduced number of audio signal components for analysis (This is nothing more than resampling the audio signal As noted many different sampling rates are used, devices cannot be connected when their sampling rates differ… For example a 44.1kHz signal can be converted to 44.056kHz by removing one sample every 23ms; page 460, fist full paragraph This could be carried to the extreme of reducing the sampling rate more drastically, such as converting from 44kHz to 22kHz by dropping every other sample).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to resample the signal as taught by Pohlmann in the system of Saunders in order to reduce the data rate in the system and thereby lowering processing requirements.

20. Consider claim 18, Pohlmann teaches the apparatus according to claim 16, wherein the at least one audio segmant component reducer comprises one or both of: a low pass filter that prevents transmission of components of the audio signal having a frequency greater than a predetermined frequency; and a decimator (This is nothing more than resampling the audio signal As noted many different sampling rates are used, devices cannot be connected when their sampling rates differ… For example a 44.1kHz signal can be converted to 44.056kHz by removing one sample every 23ms; page 460, fist full paragraph This could be carried to the extreme of reducing the

sampling rate more drastically, such as converting from 44kHz to 22kHz by dropping every other sample)

21. Consider claim 19, Pohlmann teaches the apparatus according to claim 18, wherein the decimator selecting a reduced number of audio components for analysis comprises the decimator selecting every 1 in N audio frequency components to be transmitted and selecting the audio signal components between 1 and N to be discarded (This is nothing more than resampling the audio signal As noted many different sampling rates are used, devices cannot be connected when their sampling rates differ… For example a 44.1kHz signal can be converted to 44.056kHz by removing one sample every 23ms; page 460, fist full paragraph. This could be carried to the extreme of reducing the sampling rate more drastically, such as converting from 44kHz to 22kHz by dropping every other sample, Tzanetakis, MPEG files must be decoded before conventional techniques used, therefore frequency components are decoded to time components, then processed.).

22. Consider claim 20, Benyassine teaches the apparatus according to claim 16, further comprising at least one of an audio signal encoder (figure 1, encoder 112) and an audio signal decoder (figure 1, decoder 114).

23. Consider claim 21, Benyassine teaches the apparatus according to claim 20, further comprising a speech/music classifying device being associated with the audio

signal encoder (Figure 1, encoder 112, part of transmission side, may contain a music classifier with voice activity detector; column 3, line 62.).

24. Consider claim 22, Saunders teaches the apparatus according to claim 20, further comprising a speech/music classifying device being associated with the audio signal decoder (this is a technique for discriminating speech from music from an FM broadcast; page 993, column 1, line 2. An FM signal must be decoded before it can be classified or played or manipulated in anyway).

25. Consider claim 23, Saunders teaches the apparatus according to claim 20, further comprising a signal processor and an audio processing unit associated with the audio signal decoder (The experimental setup used a Gradient A/D unit attached to a workstation; page 995, column 1, line 38. Using data processed on the fly and tuning the radio dial at will, the classification accuracy averaged between 95 and 96%; page 995, column 1, line 43. This is a technique for discriminating speech from music from an FM broadcast; page 993, column 1, line 2. An FM signal must be decoded before it can be classified or played or manipulated in anyway).

Consider claim 24, Benyassine teaches the apparatus according to claim 20, further comprising a bitstream multiplexer associated with the audio signal decoder (signal (communication devices 102 and 106 may be cellular telephones radios, or VoIP systems; column 3 line 6-11. Cell phones and VoIP systems both used packetized

data. It is inherent that some kind of multiplexing must be employed in order to packetize the data).

26. Claim 6 is rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders in view of Tzanetakis as applied to claim 1 above, and further in view of Pohlmann.

27. Consider claim 6, Saunders teaches the method according to claim 1, further comprising selecting audio signal components prior to analyzing selected audio signal components (The first step is to measure the ZCR of the signal over a 2.4 second segment of the data; page 994, column 2, line 43. Inherent the segment is selected first), but does not specifically teach wherein said selecting audio signal components comprises passing the audio signal through a decimator, wherein every 1 in N audio signal components is transmitted and audio signal components between 1 and N are discarded.

In the same field of audio processing, Pohlmann teaches, wherein said selecting audio signal components comprises passing the audio signal through a decimator, wherein every 1 in N audio signal components is transmitted and audio signal components between 1 and N are discarded (This is nothing more than resampling the audio signal As noted many different sampling rates are used, devices cannot be connected when their sampling rates differ… For example a 44.1kHz signal can be converted to 44.056kHz by removing one sample every 23ms; page 460, fist full

paragraph This could be carried to the extreme of reducing the sampling rate more drastically, such as converting from 44kHz to 22kHz by dropping every other sample).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to resample the signal as taught by Pohlmann in the system of Saunders and Tzanetakis in order to reduce the data rate in the system and thereby lowering processing requirements.

28. Claim 8 is rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders in view of Tzanetakis as applied to claim 1 above, and further in view of Benyassine (US Patent 6,694,293) and Pohlmann (Principles of Digital Audio).

29. Consider claim 8, Saunders teaches the method according to claim 1, further comprising:

selecting a number of transmitted audio signal components for analysis (The first step is to measure the ZCR of the signal over a 2.4 second segment of the data; page 994, column 2, line 43.).

However Saunders does not specifically teach transmitting components of the audio signal having a frequency less than a predetermined frequency.

In the same field of audio analysis, Benyassine teaches transmitting an audio signal using encoder 112 of figure 1 that samples at a rate of 8000Hz; column 3, line 60.

Therefore it would have been obvious to combine the sampling of audio for transmitting of Benyassine with the classification system of Saunders and Tzanetakis in order to allow the transmissions of digital signals.

This does not say specifically that that the audio being transmitted is less than a predetermined frequency.

In the same field of audio encoding, Pohlmann teaches that sampled audio must be passed through a low pass filter at the Nyquist frequency in order to prevent distortion called aliasing; page 30, prevention section.

Therefore it would have been obvious to combine the sampling of Tzanetakis with the filtering of Pohlmann in order to prevent aliasing, and to provide a way to digitize the audio signal for analysis, coding and transmission.

30. Claims 9, 15 are rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders and in view of Tzanetakis as applied to claim 1 above and further in view of Benyassine.

31. Consider claim 9, Saunders and Tzanetakis teach the method according to claim 1, but does not teach specifically wherein classifying the audio signal occurs at a transmitting end of an audio transmission system.

However in the same field of music and speech discrimination Benyassine teaches classifying the audio signals at a transmitting end of an audio transmission

system (Figure 1, encoder 112, part of transmission side, may contain a music classifier with voice activity detector; column 3, line 62.)

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to classify the music or voice at the transmitting side of the system as taught by Benyassine in order to determine properties of the signal in order to best encode the signal for transmission (Benyassine; column 1 line 62 - column 2 line 13).

32. Consider claim 15, Saunders teaches the method according to claim 1, but does not specifically teach further comprising:

converting the audio signal from an analog signal to a digital signal;

encoding the audio signal;

packetizing the audio signal;

transmitting the audio signal;

decoding the audio signal; and

processing the audio signal, wherein processing at least comprises one of storing the audio signal and playing the audio signal.

However in the same field of music and speech discrimination Benyassine teaches converting the audio signal from an analog signal to a digital signal (figure 1, A/D converter 108);

encoding the audio signal (figure 1, encoder 112);

packetizing the audio signal (communication devices 102 and 106 may be cellular telephones radios, or VoIP systems; column 3 line 6-11. Cell phones and VoIP systems both used packetized data);

transmitting the audio signal (figure 1, signals are transmitted over communication medium 104);

decoding the audio signal (using decoder 114, figure 1); and

processing the audio signal, wherein processing at least comprises one of storing the audio signal and playing the audio signal (output of system is synthesized speech signal 120, figure 1).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to use the transmission scheme of Benyassine with the audio classification method of Saunders in order to provide an efficient way to effectively transmit audio signals (Benyassine; column 1 line 62 - column 2 line 13).

33. Claim 12 is rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders and Tzanetakis. Saunders teaches the method according to claim 1, but does not specifically teach wherein the threshold value used in the comparison is pre-determined and pre-set by a user.

However Saunders does teach Data was collected manually by listening, collecting and storing features, and labeling the segment. A variety of content was processed, including talk, commercials, and many types of music. Once the classifier

was trained, the parameters were stored and fed into the real-time feature extraction/classifier routine; page 995, column 1, line 33.

With data being collected manually, it must be entered manually, and although is not specifically the threshold, one of ordinary skill in the art that the training of the classifier by manually collecting data is changing the threshold. Therefore in fact, the user is in a way changing the threshold value is preset and determined by the user.

34. Claim 17 rejected under 35 U.S.C. 103(a) as being unpatentable over Saunders in view Saunders, Tzanetakis, Benyassine, and Pohlmann as applied to claim 16 above, and further in view of Carey.

35. Consider claim 17, Saunders, Tzanetakis ,Benyassine, and Pohlmann teaches the apparatus according to claim 16, but does not specifically teach wherein classifying the audio signal based upon comparison of the recorded result of analysis and the threshold value in the comparator further comprises:

if the recorded result of analysis is greater than the threshold value, then the audio signal is determined to be music; and

if the recorded result of analysis is less than the threshold value, then the audio signal is determined to be speech.

In the same field of speech/music discrimination, Carey teaches if the recorded result of analysis is greater than the threshold value, then the audio signal is determined

to be music (table 1 shows that the mean value of number of zero crossing (u) for music 0.18 is greater than that of speech 0.17); and

if the recorded result of analysis is less than the threshold value, then the audio signal is determined to be speech (table 1 shows the mean value of zero crossing for speech 0.17 was less that music 0.18).

Although Saunders, Tzanetakis, Benyassine, and Pohlmann uses a slightly different zero crossing analysis method than does Carey, it would have been obvious to one of ordinary skill in the art at the time of the invention to use the parameters of Carey as this method would be computationally inexpensive (Carey page 151, column 2, section 4.4).

36. Claim 27 is rejected under 35 U.S.C. 103(a) as being unpatentable over Sounders and Tzanetakis as applied to claim 1 above, and further in view of Hopf et al.

37. Consider claim 27, Sounders and Tzanetakis teaches the method of claim 1, but does not specifically teach wherein selecting audio frequency components comprises selecting audio frequency components having a frequency greater than a predetermined frequency representing an upper boundary of human speech frequency.

In the same field of speech/music discrimination, Hopf teaches selecting audio frequency components having a frequency greater than a predetermined frequency representing an upper boundary of human speech frequency (frequencies from 6-10kHz

are selected as they are above where voicing is found; column 1 line 64 - column 2 line 14.).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to combine the high frequency selection of Hopf with the system of Sounders and Tzanetakis in order to in crease the accuracy of zero crossing counter by eliminating frequency components that could lead to inaccurate results.

Conclusion

38. Applicant's amendment necessitated the new ground(s) of rejection presented in this Office action. Accordingly, **THIS ACTION IS MADE FINAL**. See MPEP § 706.07(a). Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the date of this final action.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to DOUGLAS C. GODBOLD whose telephone number is

(571)270-1451. The examiner can normally be reached on Monday-Thursday 7:00am-4:30pm Friday 7:00am-3:30pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Patrick Edouard can be reached on (571) 272-7603. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

DCG

/Patrick N. Edouard/

Supervisory Patent Examiner, Art Unit 2626